SCORPIO

PREMIUM PORTABLE MIXER-RECORDER

PRELIMINARY

USER GUIDE

SOUND DEVICES

Scorpio User Guide | Rev 1-A | 04/04/19
This document is distributed by Sound Devices, LLC in online electronic (PDF) format only. published in the USA.

This table provides the revision history and cross-reference links to "what's new" in this guide.

REV #	DATE	VERSION	DESCRIPTION
1-A	04/19	A	Preliminary

Manual Conventions

SYMBOL	DESCRIPTION
>	This symbol is used to show the order in which you select menu commands and sub-options, such as: Main Menu > Outputs indicates you press the Menu button for the Main Menu, then scroll to and select Outputs by pushing the Encoder.
[ ]	This symbol is used to convey selectable menu items.
*	This symbol is used to convey factory default settings.
+	A plus sign is used to show button or keystroke combinations. For instance, Ctrl+V means to hold the Control key down and press the V key simultaneously. This also applies to other controls, such as switches and encoders. For instance, MIC+HP turn means to slide and hold the MIC/TONE switch left while turning the Headphone (HP) encoder. METERS+SELECT means to hold the METERS button down as you press the SELECT encoder.
*	A note provides recommendations and important related information.The text for notes appears italicized.
*	A cautionary warning about a specific action that could cause harm to you, the device, or cause you to lose data. Follow the guidelines in this document or on the unit itself when handling electrical equipment. The text for cautionary notes also appears italicized and bold in a different color.

Included Accessories

PART NUMBER	DESCRIPTION
2479.000	Cordset 6' AC cable
9623.001	XL-WPTA4 power supply TA4 Connector
9244.003	Scorpio LCD cover
9772.000	Antenna, SMA connector
5529.000	Promo Sticker (white)
5537.000	Promo sticker (black)
1312.000	Dot: Red, Yellow, Blue,Green, Purple, White (8 each)

SOUND DEVICES
Post Office Box 576
E7556 State Rd. 23 and 33
Reedsburg, Wisconsin 53959 USA
www.sounddevices.com

+1 608.524.0625 main
+1 608.524.0655 fax
800.505.0625 toll free
support@sounddevices.com

Dear Sound Professionals,
Thank you very much for your interest and purchase of the Scorpio. We at Sound Devices are extremely proud of this product. We consider the Scorpio our best yet, from every aspect.

We also want to thank you for your direct contribution to this product's success. Countless conversations were shared with industry professionals regarding workflows, frustrations, wants, and needs. The knowledge obtained from these conversations drove the design and engineering of the Scorpio.

Please stay in touch. We will always be here to help, listen to feature requests, and hear about your adventures with the Scorpio.

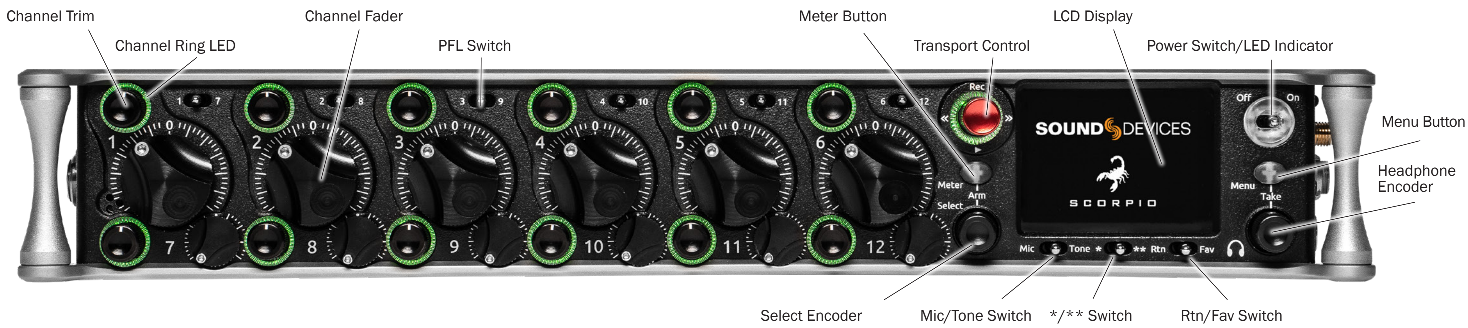
We are honored to be part of your kit.

Sincerely,
Sound Devices

Table of Contents

Panel Views

FRONT PANEL



CHANNEL TRIM Turns the channel on/off and sets the input sensitivity for the channel. To conserve power, turn off unused channels by rotating channel trim fully counter-clockwise.

CHANNEL LED RING Provides visual indication of channel signal condition, solo and mute, and whether a channel is on or off.

CHANNEL FADER Controls the audio level of the channel as it contributes to the L/R mix and any destinations selected in routing as “Post”.

PFL SWITCH Pre/Post Fade Listen selects the channel in the headphones for Pre/Post Fade Listen while simultaneously entering the channel screen. Also used for accessing virtual keyboard for channel naming and various shortcuts.

TRANSPORT CONTROLS A joystick (with its illuminated LED ring) on the front panel is used to perform various transport control functions. (see table below)

Function	Action
Record	Push up the Transport control to begin recording a new file. The LED ring illuminates red while recording is underway.
Stop	Press in the Transport control to stop recording or playback. While in standby, press and hold to display next take name.
Play	Push down on the Transport control to begin playback of the last file recorded or file currently loaded. While in playback, push down again to pause playback. The LED ring as well as the active file in the display will flash to indicate that Pause is active. Push down again to continue playback.
Rewind / Load Previous Take	While in standby, push left to load the previous take. While in playback, push and hold left to rewind. When the Scorpio is playing back or paused, moving the joystick to the left (<<) rewinds at 2x speed, then after holding for 5 seconds, it increases to 16x speed.
Fast Forward / Load Next Take	While in standby, push right to load the next take. While in playback, push and hold right to rewind. When the Scorpio is playing back or paused, moving the joystick to the right (>>) fast forwards at 2x speed, then after holding for 5 seconds, it increases to 16x speed.
Scrub	While playing or paused, press the headphone encoder to enter Scrub mode. Then rotate clockwise for fast forward or counter-clockwise for rewind speeds of 0x, 1/8x, 1/4x, 1/2x, 1x, 2x, 4x, 8x, and 16x. The audio may be heard in scrub mode up to 2x speed.

METER BUTTON Push to view and select various metering presets. Used with Select Encoder. Push again to return to Home Screen.

SELECT ENCODER
1. Push to view Outputs list, rotate and push to Select Output Screen. Push Meter Button to return to Home Screen.
2. Rotate to select track in display, push both Meter and Select at the same time to arm/disarm track. While holding the Meter Button, multiple consecutive tracks may be armed by holding in the Select Encoder and rotating.
3. Use with Meter Button to scroll through meter views then push to Select.
4. Push with Channel Select switches 1-12 for shortcut to Bus 1-10, L,R routing.
5. Menu navigation and push to Select.

MIC/TONE SWITCH Toggle slate mic and tone generator. Soft button for menus.

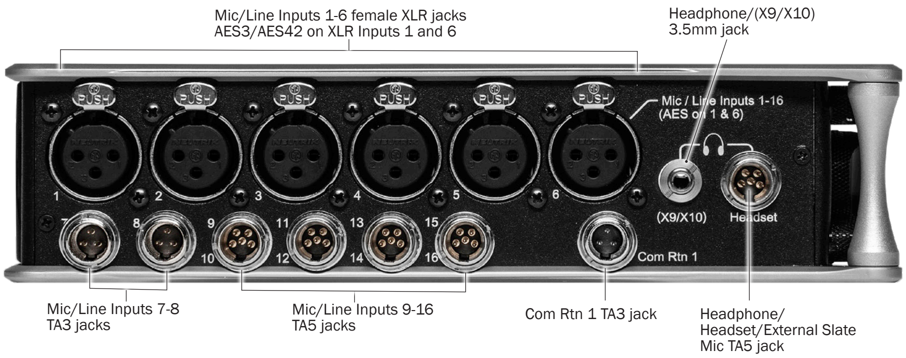
***/** SWITCH** Shortcut with PFL switch to access channels 13-24 and 25-32. Soft button for menus.

POWER SWITCH/LED INDICATOR Turns the power on and off. Switch LED ring indicates the following:
1. Power condition: green = good, orange = warning, red = shutdown imminent.
2. Flashing blue = power is off and holding timecode .
3. Continuous blue = booting up.
4. Flashing yellow = unit is off and charging L-mount batteries.
5. Continuous yellow = unit is off and both L-mount batteries are fully charged.

MENU BUTTON Push to enter the Main menu. Also used to exit menus.

HEADPHONE ENCODER
1. Rotate to control headphone volume.
2. Press to open headphone preset menu and select.
3. Menu navigation and push to select.
4. Press Menu and HP Encoder to enter Take List.

LEFT SIDE PANEL



INPUTS 1-6 FEMALE XLR JACKS Active-balanced analog microphone or line-level inputs. Inputs 1 and 6 can also accept AES3 or AES42 signal. [pin-1 = ground, pin-2 = hot (+), and pin-3 = cold (-)]

MIC/LINE INPUTS 7-8 TA3 JACKS Active-balanced analog microphone or line-level inputs. [pin-1 = ground, pin-2 = hot (+), pin 3 = cold (-)].

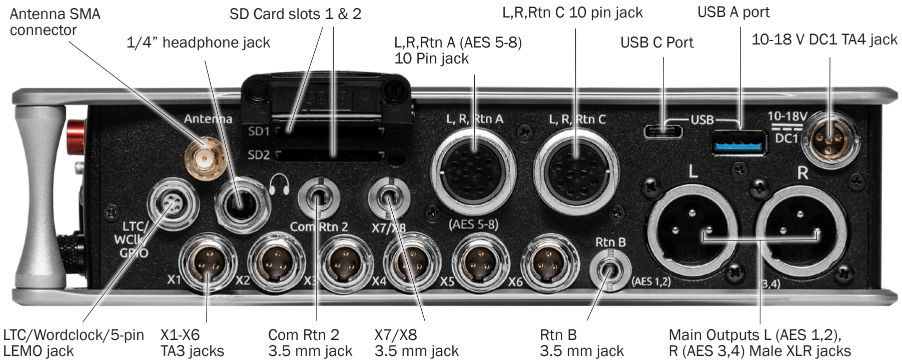
MIC/LINE INPUTS 9-16 TA5 JACKS Active-balanced analog microphone or line-level inputs. [pin-1 = ground, pin-2 = Ch.1 +, pin-3 = Ch.1 -, pin-4 = Ch.2 +, pin-5 = Ch. 2 -].

COM RTN 1 TA3 JACK Balanced connection for Com Return 1 audio input. [pin-1 = Ground, pin-2 = hot (+), pin-3 = cold (-)].

HEADPHONE/(X9/X10) 3.5 MM JACK Unbalanced output and TRS headphone output. *Warning! This output can drive headphones to potentially dangerous levels.* Routing determined in the Outputs menu. [Sleeve = ground, tip = left (X9), ring = right (X10)]

HEADPHONE/HEADSET TA5 JACK headphone and slate microphone connections [pin-1 = HP right, pin-2 = HP left, pin-3 = ground, pin-4 = Mic -, pin-5 = Mic+]

RIGHT SIDE PANEL



ANTENNA RP-SMA-MALE CONNECTOR Connects to inducted external antenna for Bluetooth LE.

SD 1 AND 2 CARD SLOTS Insert SD card media for recording. Insert label side down.

L,R,RTN A (AES 5-8) AND C 10-PIN JACKS Each connection includes a pair of outputs and a stereo unbalanced return input. Analog Output levels are selected between Line, -10, and Mic levels in Main menu > OUTPUTS section. 10-pin A outputs can be set to send AES3 digital signals (AES 5-8). Output may be sourced from L,R or any of the 10 mix busses in Main menu > OUTPUTS.

USB C PORT
1. File transfer

USB A PORT
1. USB keyboard
2. USB to SD-Remote Android app
3. USB to approved 3rd party fader controllers

10-18V DC1 TA4 JACK Accepts DC voltages from 10–18 V for powering. [pin-1- GND, pin-2- Smart Battery DATA, pin-3- Smart Battery CLOCK, pin-4- +10-18 VDC].

LTC/WORDCLOCK/5-PIN LEMO JACK Timecode I/O, Wordclock. [pin-1- GND, pin-2- LTC or WORDCLOCK IN, pin-5- LTC or WORDCLOCK OUT (Pins 2 and 5 are software selectable)].

¼" HEADPHONE JACK 1/4-inch TRS headphone output. Warning! This output can drive headphones to potentially dangerous levels. [Sleeve = ground, tip = left, ring = right].

COM RTN 2 3.5 MM JACK Balanced, 1-channel 3.5 mm female connector for Return 2 audio input. [Sleeve = ground, tip = hot, ring =cold].

X7/X8 3.5MM JACK Unbalanced stereo 3.5 mm female connector. Routing determined in the Outputs menu. [Sleeve = ground, tip = X7, ring = X8].

X1-X6 TA3 JACKS Line, -10, or Mic level selected in Main menu OUTPUTS section. Routing determined in the Outputs menu. [pin-1 = Ground, pin-2 = hot (+), pin-3 = cold (-). Float pin-3 to un-balance].

RTN B 3.5 MM JACK Unbalanced stereo 3.5 mm female connector for Return B audio input. [Sleeve = ground, tip = left, ring = right].

MAIN OUTPUTS L (AES 1,2), R (AES 3,4) XLR JACKS Analog outputs on standard 3-pin XLR-3M connectors. Analog Output levels are selected between Line, -10, and Mic levels in Main menu > OUTPUTS. Can be set to send AES3 digital signals (1,2 and 3,4 on L and R respectively) in Main menu > OUTPUTS. Routing determined in the Outputs menu. [pin-1 = Ground; pin-2 = hot (+); pin-3 = cold (-). Unbalance by floating pin-3].

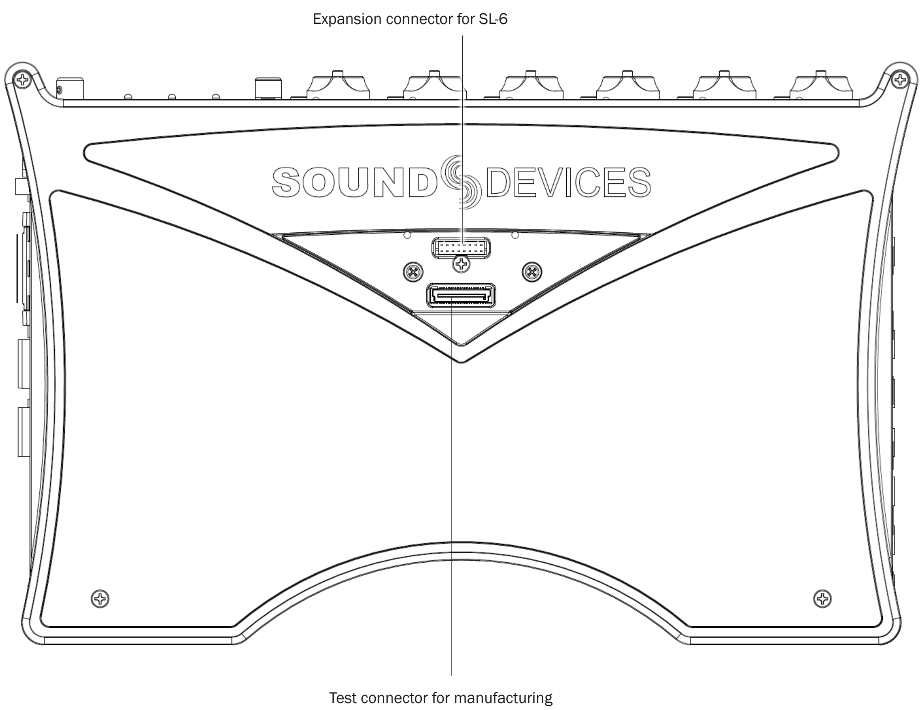
REAR SIDE PANEL

10-18V DC2 TA4 JACK Accepts DC voltages from 10–18 V for powering. [pin-1- GND, pin-2- Smart Battery DATA, pin-3- Smart Battery CLOCK, pin-4- +10-18 VDC].

BATTERY 1, BATTERY 2 DOCKING Sony L-Mount type batteries may be used. When connected to an external DC source via DC1 or DC2 the L-Mount batteries can be charged if enabled in the Power menu.

DANTE RJ45 JACKS Two-1 GbE ports serving as a Dante audio network connections. The Dante interface provides 32 inputs and 32 outputs simultaneously. Routing is defined through the Channel Source and Output menus. Dante Controller app on Mac/PC (from Audinate) needed to route and use Dante.

TOP PANEL

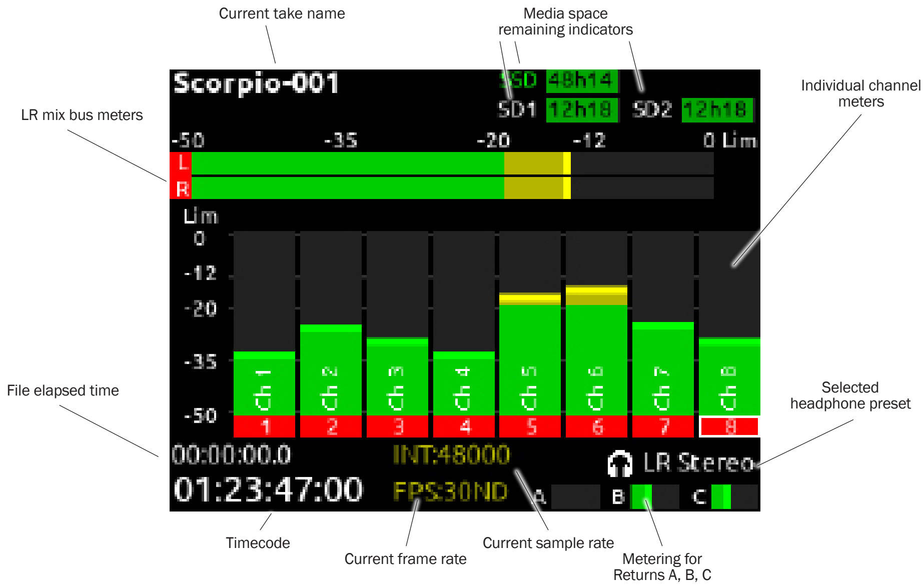


EXPANSION CONNECTOR FOR SL-6 Connection for Sound Devices SL-6 modular wireless receiver system. Routing is defined through the Channel Source menu.

TEST CONNECTOR Used during manufacturing.

Screen Views

HOME SCREEN



CURRENT TAKE NAME Shows the filename of the currently-selected take.

SSD, SD1, SD2 Indicates the used/remaining space of each form of media. The internal SSD drive has a capacity of 256 GB.

LR MIX BUS METERS WITH ARM/DISARM INDICATION
Indicates the peak and VU audio levels of the L/R mix. The L and R indicators turn red to indicate that the tracks are armed for record.

INDIVIDUAL CHANNEL METERS WITH ARM/DISARM INDICATION
Indicates the peak and VU audio levels of the individual channel. May be Pre- or Post- fade depending on Channel to Iso routing.

FILE ELAPSED/ REMAINING TIME Indicates in Hours:Minutes:Seconds:1/10ths the elapsed time of the current file. During playback, displays the elapsed and remaining time in hours, minutes and seconds.

TIMECODE Indicates current SMPTE timecode value.

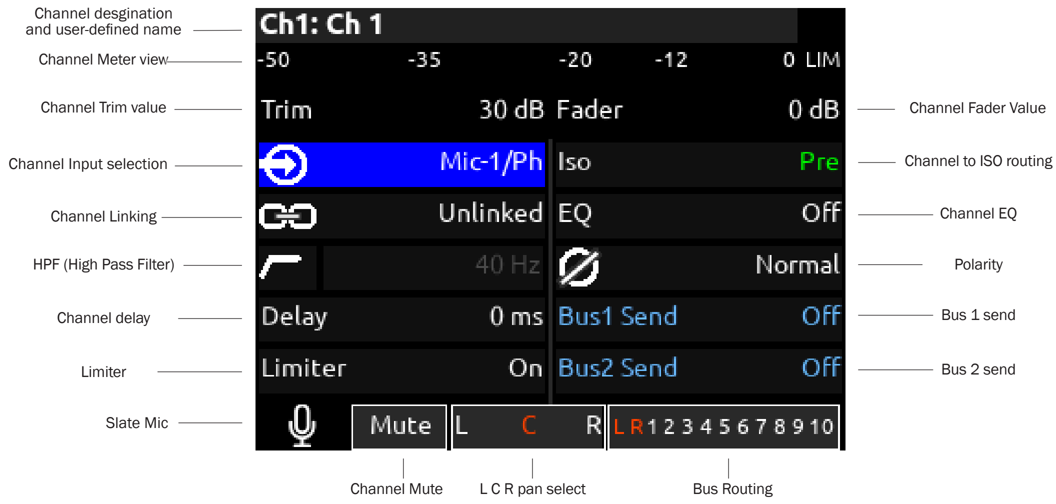
CURRENT SAMPLE RATE, FADER/TRIM LEVEL
1. Indicates current sample rate.
2. Temporarily indicates fader level of last moved fader (red text box).
3. Temporarily indicates trim level of last moved trim (green text box).

CURRENT TIMECODE FRAME RATE, FADER/TRIM LEVEL
1. Indicates current frame rate.
2. Temporarily indicates fader level of last moved fader (red text box).
3. Temporarily indicates trim level of last moved trim (green text box).

SELECTED HEADPHONE PRESET Indicates the currently-selected headphone preset.

METERING FOR RETURNS A, B AND C Indicates audio level for the returns.

CHANNEL SCREEN



CHANNEL DESIGNATION AND USER-DEFINED NAME Indicates the mixer channel designation and the user-defined name. Both are overlaid onto the channel audio meter. When in a Channel Screen, hold the PFL Switch for about 0.5 s to enter the virtual keyboard and enter a user-defined name for the channel.

CHANNEL METER VIEW Indicates the audio level of the channel. Metering follows ISO Routing selection, Pre- or Post-fade.

CHANNEL TRIM VALUE Indicates the gain of the channel trim control. The gain range depends on the type of input selected.
Mic: +12 to +76 dB
Line: -14 to +36 dB
SL-6: -20 to +50 dB
Dante: -10 to +20 dB
AES3: -10 to +20 dB
AES42: 0 to +70 dB
Returns: -20 to +30 dB

CHANNEL FADER VALUE Indicates the level of the channel fader control, continuously-variable from Off to +16dB.

CHANNEL INPUT SELECTION Indicates which physical audio input is feeding the channel.

ISO (CHANNEL->ISO) ROUTING Indicates where the isolated track's audio is tapped from in the audio chain. Pre-fade or Post-fade.

CHANNEL LINKING Indicates the current linking status. The linking options are Unlinked, adjacent channels (eg. 1,2) and adjacent channels Mid Side (eg. 1-2MS). Linked parameters are: trims, faders, HPF, delay, limiter, mute, ISO, Bus Send 1 and Bus Send 2. Stereo panning is 1 to L and 2 to R. For MS linking, the pan becomes a balance control between M and S.

CHANNEL EQ Indicates the EQ position in the audio chain. Pre(fade) or Post(fade). Select to enter Channel EQ screen.

HPF (HIGH PASS FILTER)
Indicates on/off status where green icon and white value = "On" and grey icon and value = "Off". The HPF frequency is variable in 10 Hz steps from 10 Hz to 320 Hz.

POLARITY REVERSE Indicates polarity status. Green icon = polarity reversed, white icon = polarity normal.

CHANNEL INPUT DELAY Indicates input delay time. The input delay is continuously-variable in milliseconds from 0-50 ms.

BUS 1 SEND Sends channel's audio to Bus 1. The level is continuously-variable from Off to +16 dB.

CHANNEL LIMITER Indicates on/off status of channel limiter.

BUS 2 SEND Sends channel's audio to Bus 2. The level is continuously-variable from Off to +16 dB.

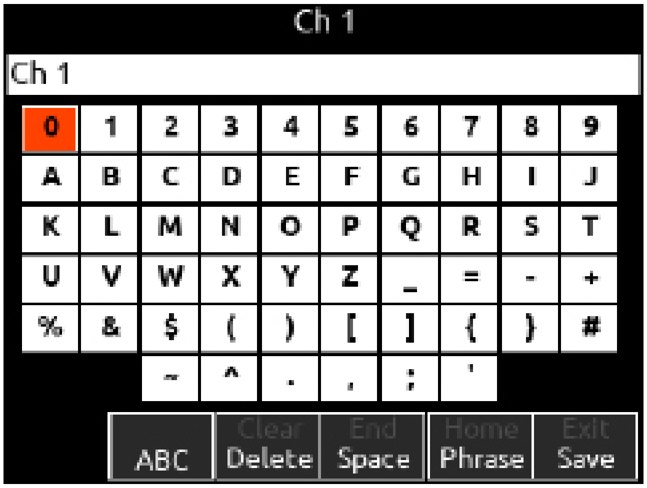
MUTE Indicates mute status of channel. Blue icon = muted. Toggle mute on/off with the "Tone" switch.

L C R SELECT Indicates the stereo pan position of the channel's contribution to the L/R mix. Orange = selected. Use the */** switch to select. Hold */** switch and rotate select encoder for continuous panning positioning. Alternatively, press and hold Select encoder, then use */** switch to pan continuously.

BUS ROUTING Indicates channel to bus routing. Green = pre-fade, Orange = post-fade, Light Blue = via Bus Send, White = No route. Use the Rtn/Fav switch to access the selected channel's bus sends view. Use the HP encoder to scroll and select the bus type (Pre, Post or Send) and set the gain for the channel's contribution to the bus. Use the Menu button to return to the channel screen.

Virtual Keyboard

Action	Function
Rotate HP	Scrolls orange highlight through the keyboard characters.
Press HP	Inserts the highlighted character in text field.
‘abc’ switch	Quick flick toggles between A-Z and a-z in keyboard.
Hold ‘abc’ switch	Momentary selection of other case.
Delete	Deletes character to the left of flashing cursor.
Hold Delete	Repeatedly deletes characters to the left of flashing cursor.
Space	Inserts space at the flashing cursor position.
Hold Space	Repeatedly inserts spaces.
Save switch	Saves text and exits screen.
Rotate Select	Moves the cursor to the left or right in the text field.
Quick Press Select	Switches to the the Shifted functions: Clear, End, Home, Exit. When shifted functions are active, their text changes to white and the non-shifted functions change to gray.
Clear	Clears text from the text edit field.
End/Home	Moves cursor to end/start of text.
Exit	Exits screen without saving text edits.



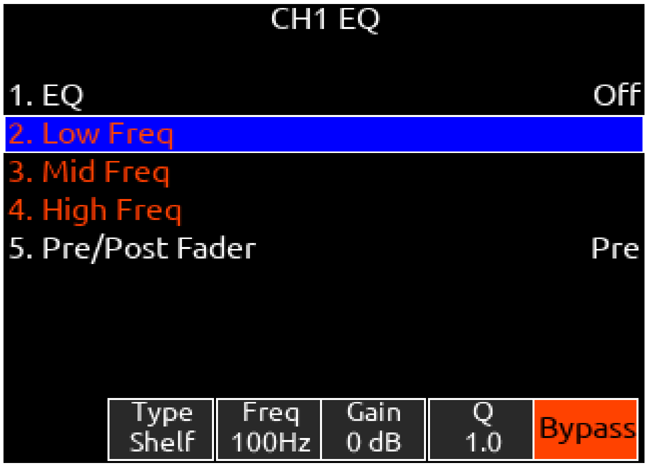
1. EQ Selects channel EQ state. [Off*, On].

2. LOW FREQ Selects low frequency EQ filter parameters.
a. Type- Indicates Shelf or Peaking filter [Shelf, Peak].
b. Freq- Indicates frequency of the filter, continuously-variable from 20 Hz to 20 kHz (100Hz*).
c. Gain- Indicates gain of the filter, continuously-variable from -15dB to +15 dB in 1 dB increments (0 dB*).
d. Q- Indicates “Q” or bandwidth of the filter, continuously-variable from .5 - 10 in .1 increments (1.0*).
e. Bypass- Indicates state of the filter [Bypass (orange fill)]*.

3. MID FREQ Selects mid frequency EQ filter parameters.
a. Freq- Indicates frequency of the filter, continuously-variable from 200-20 kHz in 10 Hz increments (5000Hz*).
b. Gain- Indicates gain of the filter, continuously-variable from -15 dB to +15dB in 1dB increments (0dB*).
c. Q- Indicates “Q” or bandwidth of the filter, continuously-variable from .5-10 in .1 increments (1.0*).
d. Bypass- Indicates state of the filter [Bypass(orange fill)]*.

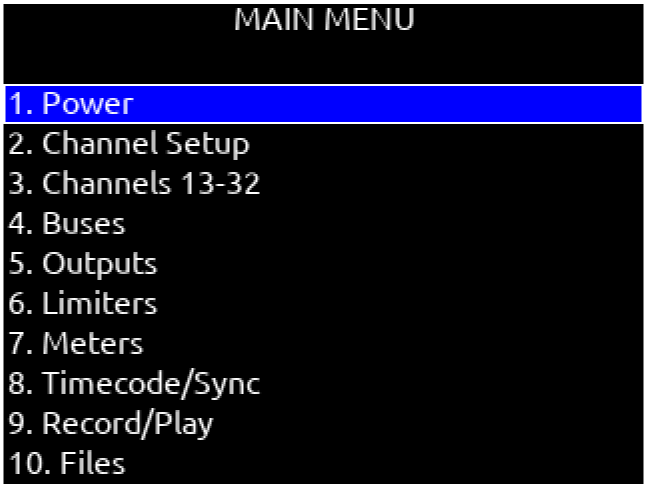
4. HI FREQ Selects high frequency EQ filter parameters.
a. Type- Indicates Shelf or Peaking filter [Peak, Shelf].
b. Freq- Indicates frequency of the filter, continuously-variable from 20 Hz to 20kHz. (100 Hz*).
c. Gain- Indicates gain of the filter, continuously-variable from -15dB to +15dB in 1dB increments. (0 dB*)
d. Q- Indicates “Q” or bandwidth of the filter, continuously-variable from .5 - 10 in .1 increments. (1.0*)
e. Bypass- Indicates state of the filter. [Bypass (orange fill)]*

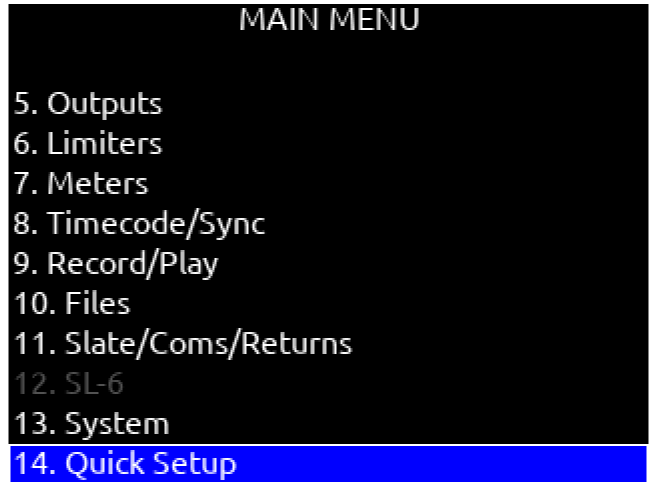
5. PRE/POST-FADER Indicates where the EQ is inserted into the audio chain. Pre-fade or Post-fade [Pre*, Post] *Note: EQ will apply to bus sends when applied Pre-fade only.



Menus

MAIN MENU





Power

Allows configuration of various power settings.

POWER SOURCE ICONS “Batt1, Batt2, DC1, DC2, TC Batt, SL-6 DC, SL-6 NP1” Indicates the power condition of each of the power sources. Green = normal, Yellow = below normal, Orange = low, Red = warning

1. DC1 REF Allows proper power level indicator calibration based upon the type of DC power source used. [12V DC*, 14 V Li-Ion, 12 V Lead Acid, NiMH, Expanded NiMH, Full Range (10-18V), Smart Battery].

2. DC2 REF Allows proper power level indicator calibration based upon the type of DC power source used. [12V DC*, 14 V Li-Ion, 12 V Lead Acid, NiMH, Expanded NiMH, Full Range (10-18 V), Smart Battery].

3. DC LOSS Selects how the unit should operate when DC power is lost. [Switch to Next Supply*, Turn Off].

4. BATT CHARGING Enable/Disable battery charging when connected to an external DC source. [Enable, Disable*].

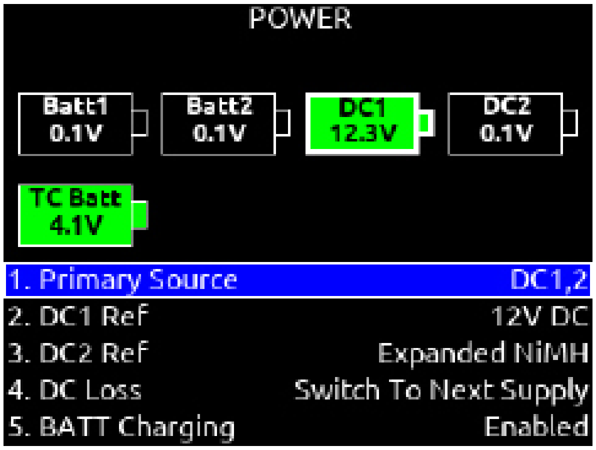
5. SL-6 PRIMARY SOURCE Selects the primary power source for the SL-6. [NP1*, DC Input].

6. SL-6 NP1 REF Allows proper power level indicator calibration based upon the type of battery used. [14 V Li-Ion*, NiMH].

7. SL-6 DC REF Allows proper power level indicator calibration based upon the type of DC power source used. [12 V DC*, 14 V Li-Ion, 12 V Lead Acid, NiMH, Expanded NiMH, Full Range (10-18 V)].

8. SL-6 POWER OUTPUTS Enables or disables onboard power outputs [Off*, On].

**Note: All SL-6 related items are only shown in the display when an SL-6 is connected to the Scorpio.*



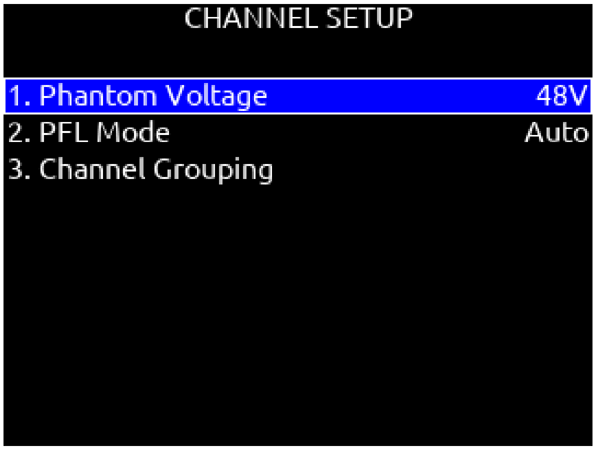
Channel Setup

1. PHANTOM VOLTAGE Selects phantom power voltage for all inputs. [12 V, 48 V*].

2. PFL MODE Selects the source of the PFL feed. [Auto* Pre-fade, Post-fade] Auto = pre-fade if channel is routed to ISO track pre-fade, post-fade if channel is routed to the ISO track post-fade.

3. CHANNEL GROUPING Selects grouping of faders, record arming, and mutes across channels. The lowest channel number in the group controls the other channels grouped. Four channel groups are possible, channels grouped can only be assigned to one group.
a. Group 1 [1-16]
b. Group 2 [1-16]
c. Group 3 [1-16]
d. Group 4 [1-16]

CHANNELS 13-32 Selects channel screen for channels 13-32 [Channel 13-32].



Buses

Selects routing for Buses L,R and 1-10.

1. BUS METER Audio level meter for the selected bus.

2. LINK *-* Selects linking for two even-to-odd numbered adjacent buses. Links bus Gain, Mute Coms, and Mute All functions.

3. ISO Any ISO channel contributes to Bus mix. Green fill in text box = Pre-fade, Orange fill in text box = Post-fade, Blue fill in text box = Send (eg. Bus routing screen with Channel 1 box with blue fill = Channel 1 sending to Bus 1 via Aux 1 send) [1-32].

4. BUS [L,R, 1,2 (available on buses 3-10).

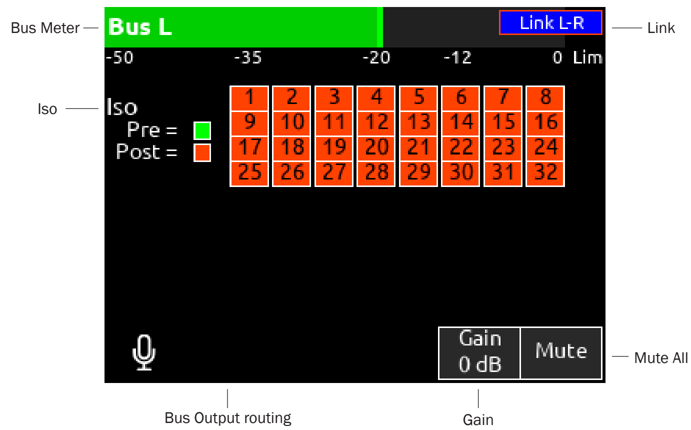
5. RTN COM [Send 1, Send 2, RTN 1, RTN 2 (not available on L,R busses)].

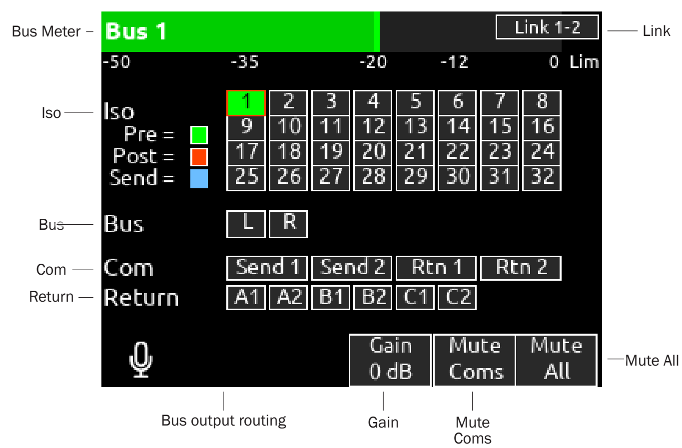
6. RETURN [A1, A2, B1, B2, C1, C2 (not available on L,R buses)].

7. GAIN Use ** toggle to select and adjust selected bus gain in 1 dB increments. [Off-16dB].

8. MUTE COMS Selects muting of Coms 1, 2 sends and returns.

9. MUTE ALL Indicates mute status of bus. Blue icon = muted. Toggle Mute All On/Off with the “Fav” toggle.





Outputs

1. LR, X1-X10 Output Routing

Selects routing for L,R and X1-X10 outputs [L Out, R Out, X1, X2, X3, X4, X5, X6, X7, X8, X9 and X10 Out] **Only a single source can be routed to an Output. If multiple sources need to be routed, use a Bus**

A. ISO Selected source will contribute to the Output. Green = Pre-fade, Orange = Post-fade [1-32].

B. BUS [L,R, 1-10, HP-L, HP-R].

C. COM [Send 1, Send 2, RTN 1, RTN 2].

D. RETURN COM [A1, A2, B1, B2, C1, C2].

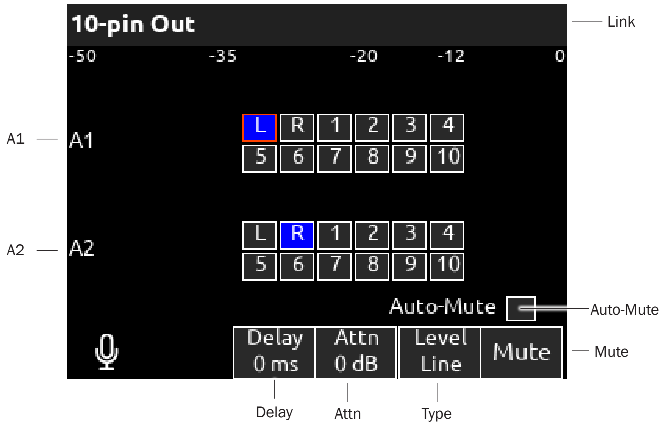
E. AUTO-MUTE Selects automatic muting of the output when in Stop mode. Record and Playback modes are not muted.

F. DELAY The output delay is continuously-variable in milliseconds from 0-500 ms.

G. GAIN Selects amount of attenuation applied to the output. Toggle the ** to select [0 dB to -50 dB and -inf].

H. LEVEL Selects output level type. [Line, -10, Mic].

J. MUTE Indicates mute status of output. Orange = muted. Toggle Mute On/Off with the “Fav” toggle.



2. 10-Pin A Out Routing

Selects routing for 10-Pin A outputs.

A. A1 Selects mix bus for A1 output [L,R, 1-10].

B. A2 Selects mix bus for A2 output [L,R, 1-10].

C. AUTO-MUTE Selects automatic muting of the output when in Stop mode. Record and Playback modes are not muted.

D. DELAY The output delay is continuously-variable in milliseconds from 0-500 ms.

E. GAIN Selects amount of attenuation applied to the output. Toggle the ** to select [0 dB to -50 dB and -inf].

F. LEVEL Selects output level type. AES option is available for L, R, 10-pin A [Line, -10, Mic].

G. MUTE Indicates mute status of bus. Blue icon = muted. Toggle Mute On/Off with the “Fav” toggle.

3. 10-Pin C Level

Selects the output level type [Line*, -10, Mic].

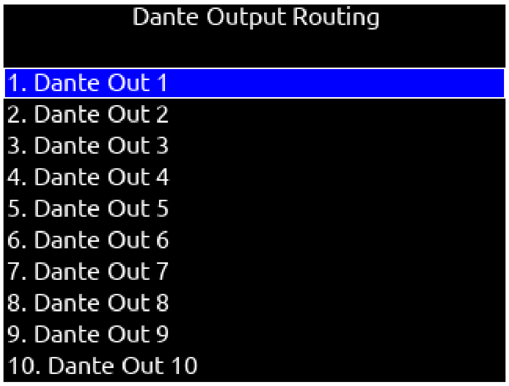
4. Dante

Selects routing for Dante outputs.

A. ISO Any source selected will be routed to the selected Dante output. Green fill in text box = Pre-fade, Orange fill in text box = Post-fade [1-32]

B. BUS [L,R, 1-10].

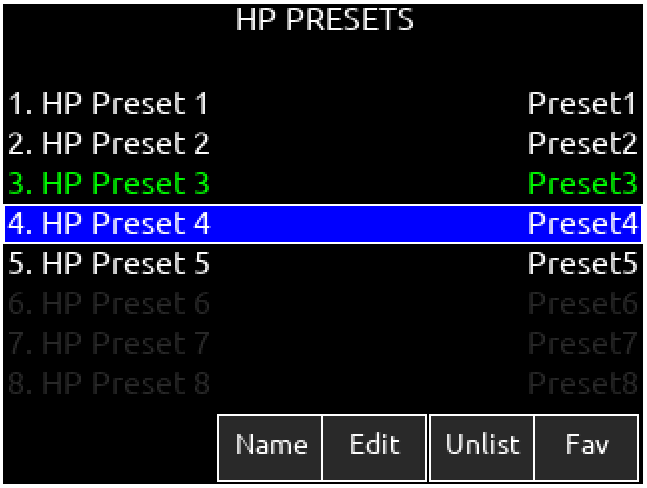
C. OUTPUT All sources are selected post-delay [L,R, X1-X10].



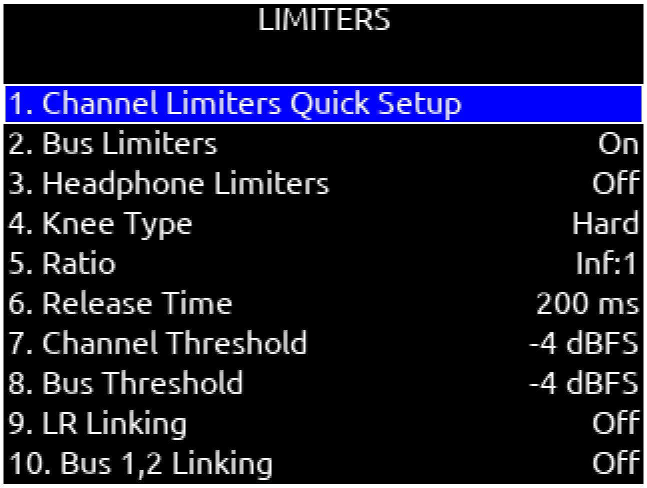
5. HP Presets

Selects the list of headphone presets available and allows for editing and creation.

Function	Description
Name	Displays virtual keyboard and allows for naming of the headphone preset.
Edit	Allows selection of routed sources to both HP Left and HP Right. Select HP LEFT or RIGHT and then select desired source. i. Iso- Any source selected will be routed to the selected HP output. Green = Pre-fade, Orange = Post-fade. [1-32] ii. Bus- [L,R, 1-10] iii. Return Com- [Send 1, Send 2, RTN 1, RTN 2] iv. Return- [A1, A2, B1, B2, C1, C2]
Mono	Selects monophonic monitoring of selected HP-L/HP-R sources.
MS	Selects monophonic monitoring of selected HP-L/HP-R sources.
Unlist	Deselects a preset in the list preventing it from being listed in the HP Preset menu (Press HP encoder in Home Screen).
List	Selects a preset in the list allowing it to be listed in the HP Preset menu (Press HP encoder in Home Screen).
Fav	Selects a favorite preset. The name turns green when selected. The “Fav” switch recalls this HP preset when in the Home Screen.



Limiters



CHANNEL LIMITERS QUICK SETUP Selects the channel limiters on/off status globally. [All On*, All Off]

BUS LIMITERS Selects the bus limiters on/off status globally. [All On*, All Off]

HEADPHONE LIMITERS Selects the headphone limiters on/off status. [On, Off*]

KNEE TYPE Selects the knee type of the limiter. [Hard*, Soft]

RATIO Selects the ratio of the limiter. [Inf:1*, 20:1]

RELEASE TIME Selects the release time of the limiters in 10 ms increments. 200 ms* [50-1000 ms]

CHANNEL THRESHOLD Selects the threshold at which the channel limiters activate. -4 dBFS Default [-2 to -12 dBFS]

BUS THRESHOLD Selects the threshold at which the bus limiters activate. -4 dBFS Default [-2 to -16 dBFS]

LR LINKING Selects the linking of the L and R limiters. [On*, Off]

BUS 1,2 LINKING- Selects the linking of the bus 1 and 2 limiters. [On, Off*]

BUS 3,4 LINKING Selects the linking of the bus 3 and 4 limiters. [On, Off*]

BUS 5,6 LINKING Selects the linking of the bus 5 and 6 limiters. [On, Off*]

BUS 7,8 LINKING Selects the linking of the bus 7 and 8 limiters. [On, Off*]

BUS 9,10 LINKING Selects the linking of the bus 9 and 10 limiters. [On, Off*]

Meters

Selected Preset

METER PRESETS 1-12

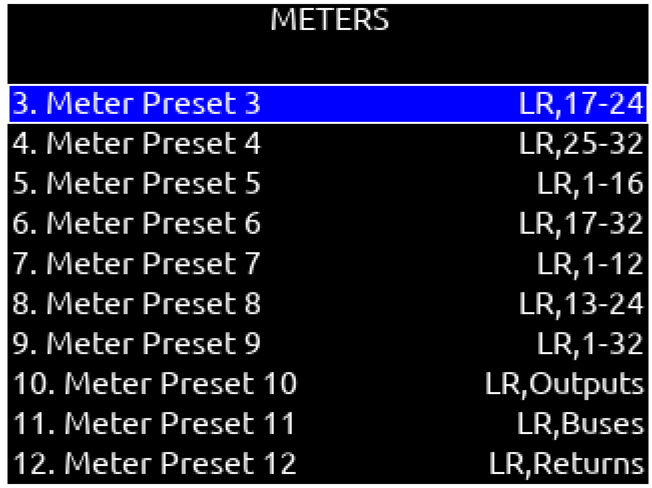
A. PEAK HOLD TIME Selects the peak hold time for the meter preset. [Off, 1*-5s., Infinity]

B. METER RANGE Selects the range of the meters from bottom to top of scale. [50 dB*, 40 dB, 20 dB]

C. METER VIEW Selects the meters to be viewed in the current preset. [LR,1-8, LR,9-16, LR,17-24, LR,25-32, LR,1-16, LR,17-32, LR,1-12, LR,13-24, LR,1-32, LR,Outputs, LR,Buses, LR,Returns]

D. TRACK NAMES Selects display of track name in meters. [Enabled*, Disabled]

E. GRAY METERS Selects gray meter when record disarmed. [When disarmed*,Off]



Timecode

TIMECODE MODE Selects the timecode mode of operation. [Off, Record Run, Free Run*, Free Run Auto Mute, Free Run Jam Once, 24 Hour Run (ToD), 24 Hour Run Auto Mute, Ext TC, Ext TC - Auto Record, Ext TC Continuous, Ext TC Cont. - Auto Record]

FRAME RATE Selects the current frame rate. [23.98, 24, 25, 29.97 ND, 29.97 DF, 30 ND*, 30 DF]

HOLD OFF Selects the amount of time incoming Timecode needs to be valid prior to entering record when in auto-record mode. [0.0*-8.0 seconds in steps of 0.1 sec.]

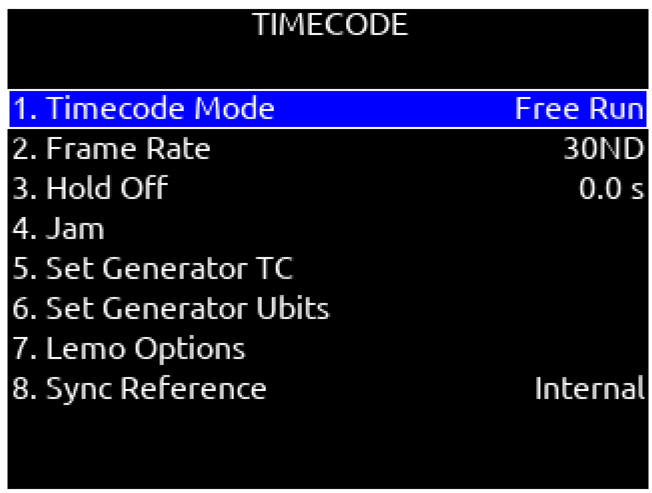
JAM Indicates the Received TC, Generator TC and the calculated difference between the two. Received and Generator UBits are shown. Jamming to external TC and UBits is supported.
a. Jam TC- Toggle Rtn/Fav switch to jam to external TC.

SET GENERATOR TC Provides the ability to start rolling internal TC from a manually entered value in the format of HH:MM:SS:ff

SET GENERATOR UBITS Provides UBits manual and automatic entry. [U=User entered UU:UU:UU:UU*, mm:dd:yy:UU, dd:mm:yy:UU, Use External] Use Rtn/Fav toggle to exit.

LEMO OPTIONS Selects pin-2 and pin-5 options for TC Lemo connector.
a. Pin-2 - [TC In*, WCK In, WCK Out]
b. Pin-5 - [TC Out, WCK Out]

SYNC REFERENCE Selects current sync reference. [Internal*, Word Clock, LTC In]



Record/Play

SAMPLE RATE Selects the current sample rate. [44100, 47952, 48000*, 48048, 96000, 192000]

BIT DEPTH Selects the current bit depth. [16, 24*]

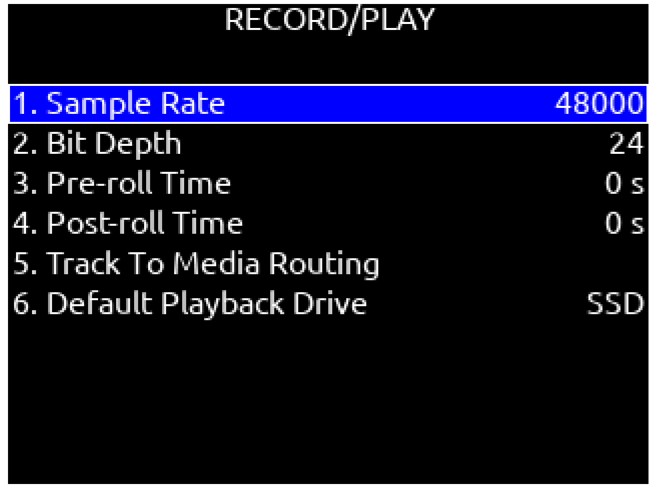
PRE-ROLL TIME Selects the amount of Pre-roll recording. Adjustable in 1 second increments. *2 s. [0-5 s.]

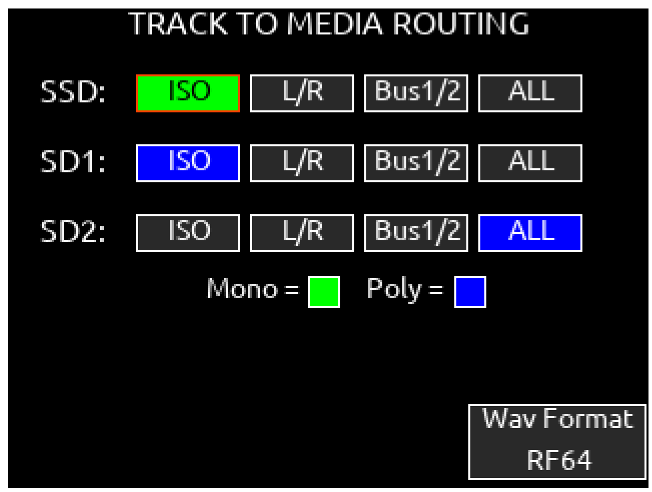
POST-ROLL TIME Selects the amount of Post-roll recording. Adjustable in 1 second increments. [0-10 s.] If a recording is stopped prematurely, press record within the post-roll time. The machine will continue to record into the original file. Useful for when directors call ‘cut’ prematurely. During the post roll period, the transport joystick ring LED shows orange. Pressing stop again during the post-roll period cancels the post roll and stops recording.

TRACK TO MEDIA MENU Selects the sources for each recording media as well as the .wav file type recorded. Tracks may be routed to media to be recorded as Mono or Poly files. (Green fill in text box= Mono file, Blue fill in text box= Poly file)
a. SSD- [ISO, L/R, Bus1/2, ALL]
b. SD1- [ISO, L/R, Bus1/2, ALL]
c. SD2- [ISO, L/R, Bus1/2, ALL]

** Note: Up to 36 track recording supported with sampling rates 44.1-96 kHz. Up to 18 track recording at 192 kHz.*
***Note: Monophonic file recording up to 48.048 kHz.*

DEFAULT PLAYBACK DRIVE Selects the drive for playback. [SSD, SD1, SD2]





Slate/Coms/Returns

SLATE/COM MIC SOURCE Selects the slate and com mic source. [Off, Int Mic*, Ext Mic, Ext 12 V Mic]

SLATE/COM MIC GAIN Selects the gain for the slate/com mic. [0-20 dB in 1 dB steps for the internal mic, 0-60 dB in 1 dB steps for the external mic].

SLATE ROUTING Selects the destination(s) for the slate mic.
a. Track- [1-32]
b. Output- [L,R, X1-X10]
c. Bus- [L,R, 1-10] [Mute Program, Unmute Program, Duck Program]
d. HP- [HP-L, HP-R] [Mute Program, Unmute Program, Duck Program]
e. Duck By: [0- -40 dB]

COM SEND 1 ROUTING Selects the destination(s) for Com Send 1.
a. Output- [L,R, X1-X10, A1,A2]
b. Bus- [L,R, X1-X10, Mute Program]
c. HP- [HP-L, HP-R, Mute Program]
d. Duck By: [0- -40 dB]

COM SEND 2 ROUTING Selects the destination(s) for Com Send 2.
a. Output- [L,R, X1-X10, A1,A2]
b. Bus- [L,R, X1-X10, Mute Program]
c. HP- [HP-L, HP-R, Mute Program]
d. Duck By: [0- -40 dB]

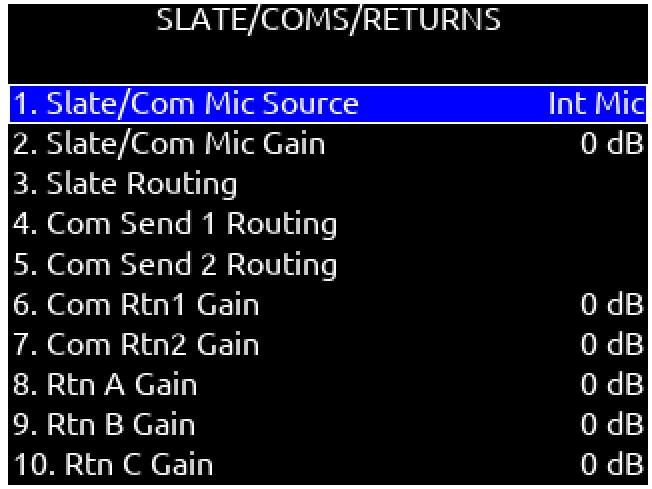
COM RTN1 GAIN Selects the gain for COM RTN 1 in 1 dB increments. [0-30 dB]

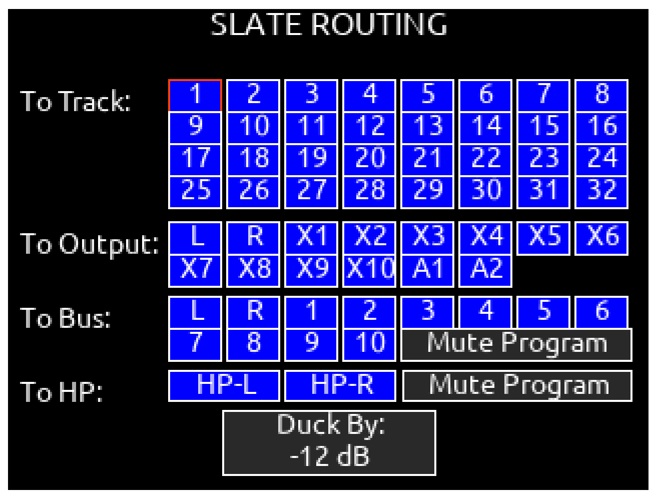
COM RTN2 GAIN Selects the gain for COM RTN 2 in 1 dB increments. [0-30 dB]

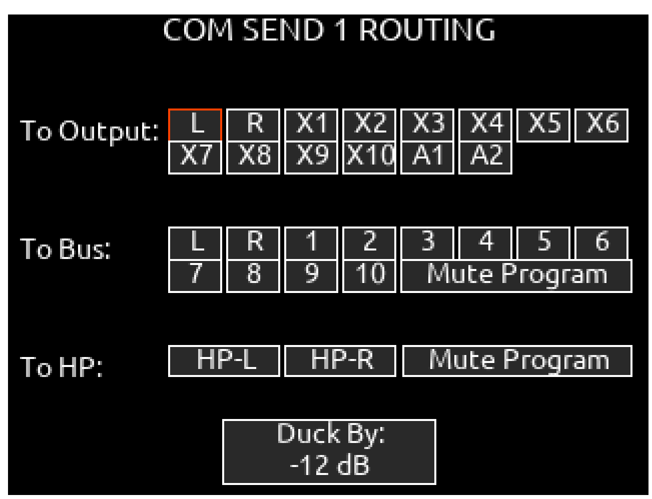
RTN A GAIN Selects the gain for RTN A in 1 dB increments. [0-30 dB]

RTN B GAIN Selects the gain for RTN B in 1 dB increments. [0-30 dB]

RTN C GAIN Selects the gain for RTN C in 1 dB increments. [0-30 dB]







Files

USB FILE TRANSFER Enters USB file transfer mode. Files may be transferred between a Mac or PC and Scorpio via USB-C port.
**Note: when in USB file transfer mode, all other audio playback and record functions are suspended.*

TAKE LIST Enters the Take List. The Take List shows a running list of recorded takes in chronological order, most recent at the top. Various details of each take are indicated on the right side of the display: TC (timecode), Duration, Media, Folder, Scene, Take, Date, and Notes. From this list, takes may be selected for metadata editing by using the Rtn/Fav toggle.

FILE LIST Enters the File List. The File List displays the Scorpio's internal SSD and SD cards and their contents. Various details of each drive, folder, and WAV file are indicated on the right side of the display: TC, FPS:, duration, format, tracks, date, time, size.

SOUND REPORT INFO Selects the various content for each field of a sound report.

SCENE INCREMENT MODE Defines whether a scene name shall be incremented numerically or alphabetically when the scene increment shortcut is used.

TAKE RESET MODE Selects when a take number shall reset to 1 when the scene or daily folder changes.

ERASE/FORMAT SSD Select to erase/format the internal SSD.

ERASE/FORMAT SD1 Select to erase/format SD1.

ERASE/FORMAT SD2 Select to erase/format SD2.

SL-6

**Note: The SL-6 menu is only available when an SL-6 is connected. Power must be connected to the SL-6 via the SL-6 power connections for use with Scorpio.*

RECEIVER OVERVIEW Information including RF level, Transmitter battery level, channel frequency, and audio level.

RECEIVER SLOT POWER Selects powering on and off of each SL-6 receiver slot.

ANTENNA A POWER (BIAS) Selects 12 VDC power for active antenna use.

ANTENNA B POWER (BIAS) Selects 12 VDC power for active antenna use.

ANTENNA FILTER Selects the SL-6 front-end filter bandwidth.
a. Wideband.
b. 470-700MHZ.
c. 470-590MHZ.
d. 580-700MhZ.

System

TONE SETUP Selects the level, frequency, and routing of the internal tone generator.
a. Level- Selects the level of the tone generator from -20 - 0 dBFS in 1 dB increments. [-20 - 0 dBFS]
b. Frequency- Selects the frequency of the tone from 100 to 10 kHz in 10 Hz steps. [100-10 kHz]
c. Track- [1-32]
d. Output- [L,R, X1-X10]
e. Bus- [L,R, 1-10] [L-ident]

NOTIFICATION BELLS Selects settings for the notification bells.
a. To HP- Routes notification bell tones to the headphones. [HP-L, HP-R]
b. To Bus- Routes notification bell tones to the buses. [L,R, 1-10]
c. When...- Selects when the notification bell tones are used. [Rec/Stop, Space Low, Power Low, Warning Popup]
d. Level- Selects the level at which the notification bell tones will be played in 1 dB increments. [Muted, -60 to -12dBFS]

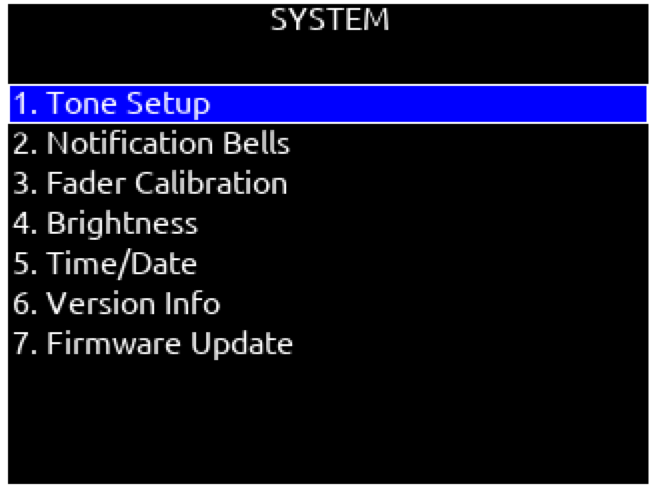
FADER CALIBRATION Selects the option to manually calibrate all faders.

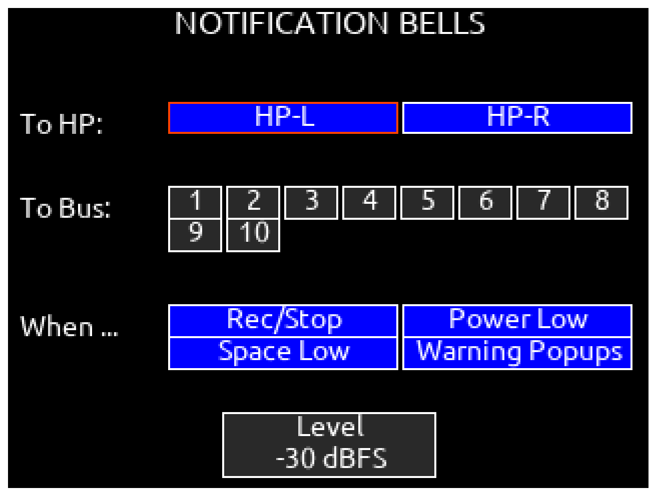
BRIGHTNESS Selects the brightness of the LED display and front panel LEDs.
a. LED Brightness- Selects the front panel LED brightness in 10% steps. [10%-100%]
b. Selects the front panel LCD display brightness in 10% steps. [10%-100%]

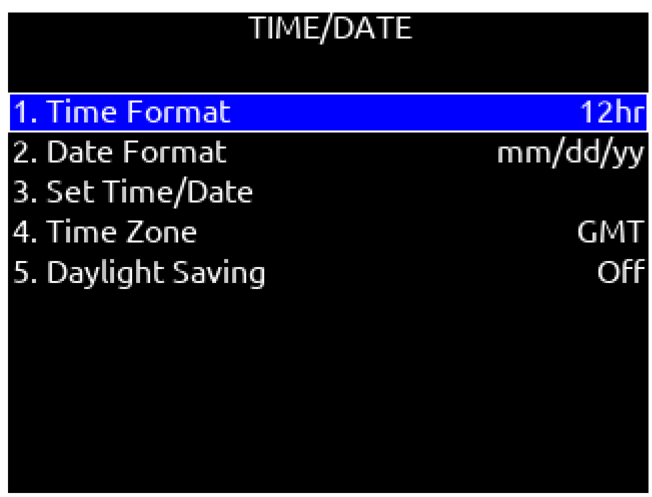
TIME/DATE Selects the current date and time.
a. Time Format- [12*, 24 hr]
b. Date Format- [mm/dd/yy*, dd/mm/yy, yy/mm/dd]
c. Set Time/Date- Selects the current date and time
d. Time Zone- [-12 to +13 hours GMT]
e. Daylight Saving- [On, Off*]

VERSION INFO Indicates current firmware version

FIRMWARE UPDATE Selects any *.prg update files present on any media.





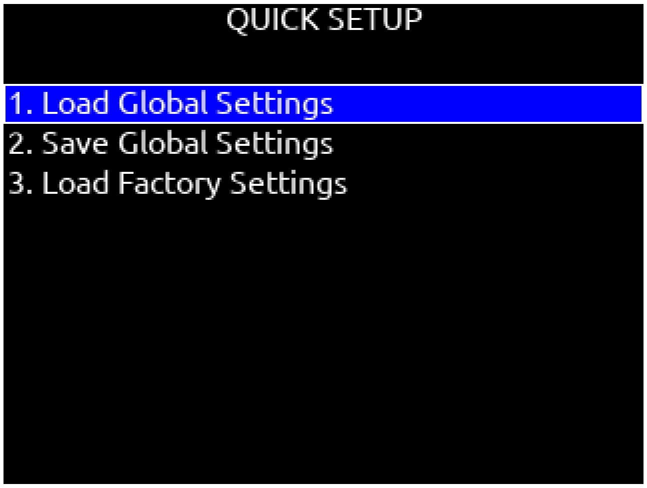


Quick Setup

LOAD GLOBAL SETTINGS Selects a saved settings file for loading. [User-saved Global settings]

SAVE GLOBAL SETTINGS Saves Global settings to various destinations. [SSD Drive (internal), INT1-4 (internal), SD1 and SD2]

LOAD FACTORY SETTINGS Selects factory settings to be loaded for entire unit.



Front Panel Shortcuts

quick = quickly flick toggle for < 0.5 seconds

Channel Screen access and PFLs	
Hold PFL (1-12)	PFL channel (1-12) and stay on current screen
PFL (1-12) quick	PFL channel (1-12) and access channel screen
Hold * then PFL (1-12)	PFL channel (13-24) and stay on current screen
Hold * then PFL (1-12) quick	PFL channel (13-24) and access channel screen
Hold ** then PFL (1-8)	PFL(25-32) and stay on current screen
Hold ** then PFL (1-8) quick	PFL channel (25-32) and access channel screen

Meter View, Menu, Output, and HP Preset Quick Access	
Hold Meter then PFL (1-12)	Accesses METER views 1-12 respectivly
Hold Select then PFL (1-12)	Accesses BUS screens L,R, X1-X10 respectivly
Hold HP then PFL (1-12)	Acesses HP presets 1-12 respectively
Press Meter	Displays meter view popup list. Then scroll using Select or PFL 1-12 to select favorite menu 1-12
Press Select	Displays outputs popup list. Then scroll using select to select output screen L,R,X1-X10
Press HP	Displays HP preset popup list. The scroll using HP to select HP preset 1-12

In Channel Screen	
Rotate then press Select	Navigate to, then enter a channel screen field
*/**	3-position pan adjustment LCR
Hold Select then */**	Continuous pan adjustment (L10 to R10)
Hold */** then rotate Select	Continuous pan adjustment (L10 to R10)
Hold PFL > 0.5 seconds	Shortcut to channel naming

Other Button Shortcuts	
Meter + Select or Select + Meter	Arm or disarm Slected track
Menu + HP or HP + Menu	Accesses Take List
HP + Play	Switches the playback drive between SSD, SD1, SD2
Hold Meter then Select	Arms or disarms multiple individual tracks at one time
Hold Meter then press and rotate Select, then release Select and Meter	Arms or disarms multiple consecutive tracks at one time

Dante™

Scorpio is capable of connecting to a Dante network, receiving and sending up to 32 channels of audio simultaneously at sample rates from 44.1 kHz up to 96 kHz and 16 channels at 192 kHz. Scorpio channels 1-16 may be sourced from Dante receive channels 1-16. Scorpio channels 17-32 may be sourced from Dante receive channels 17-32. Each Dante input may be selected as a source in the channel setup menu. Each Dante output may be sourced from Isos (pre- or post-fader), Buses, and Outputs (post-delay). All network routing should be done through Audinate's Dante Controller application, found at www.audinate.com. Once the initial configuration has been performed, the Scorpio will keep its Dante configuration through power cycles. It is recommended that in most situations Scorpio is selected as "Preferred Master" under the "Clock Status" tab of Dante Controller.

USB-A

USB-A allows multiple devices to be used to control and monitor various functions of Scorpio. Should multiple devices be used simultaneously, the use of a USB-A type hub is required.

EXTERNAL CONTROLLERS

Scorpio supports select 3rd-party USB controllers via open MCU protocol.
Control and display parameters include the following:
1. Channel Name
2. Channel Faders
3. Channel Trims
4 Channel Pans
5. Channel Mutes
6. Channel Solos
7. Channel Record Arm/Disarm
8. Channel "fat channel" (Source, Limiter, HPF, EQ, Bus sends)
9. Channel bank switching
10. Bus Master Faders
11. Output Master Faders
12. Transport Controls

SD-REMOTE

SD-Remote is an Android tablet application, available in the Google Play Store. It is designed to pair with Scorpio and display parameters including the following:
1. Channel Meters
2. L/R Meters
3. Channel Name
4. Channel Solos
5. Channel Record Arm/Disarm
6. Transport Controls
7. Metadata Editing
8. Various Status Info
9. Timecode
10. Channel Mutes

SETUP PROCEDURE

1. Download and install SD-Remote from the Google Play Store
2. Connect Android tablet to Scorpio via USB-A port
3. On the Android tablet, open the quick settings drop down menu
4. Touch "USB Android System" twice to open "Use USB to" dialog box
5. Touch "Connect a MIDI device"
6. Open SD-Remote app

USB KEYBOARD

A USB keyboard connects to Scorpio via the USB-A port. The keyboard is used for metadata entry as well as the following shortcuts:

Key	Scorpio Function
ESC	"Back" or Cancel Virtual Function
F1 or Menu	Menu
F2	Take List
F3	Toggles through all meter views
Ctrl + R	Record
Ctrl + S	Stop
Spacebar	Play/Pause
Up/Down Arrows	Up/down navigation
Enter	Selects highlighted items in menus In meter views, opens the HP Preset list
Alt + Enter	Track Arm/Disarm of selected tracks

USB-C

USB-C allows for high-speed file transfer between a computer and any of the Scorpio's media.
**All other functionality is suspended in USB File Transfer mode.*

Specifications

Specifications are subject to change without prior notice.
For the latest information available on all Sound Devices products, visit our website: www.sounddevices.com.

FREQUENCY RESPONE
10 Hz to 80 kHz ± 0.5 dB (192 kHz sample rate, re 1 kHz)

THD + NOISE
0.005% max (mic in, 1 kHz, 22 Hz–22 kHz BW, trim at 20, fader at 0, -10 dBu in)

EQUIVALENT INPUT NOISE
-131 dBV (-129 dBu) max (mic in, A-weighting, 76 dB gain, 150 ohm source impedance)

PROCESSING ENGINE
Highly extensible, full FPGA-based audio processing, 3 FPGAs
Six-way ARM multiprocessor system
64-bit audio processing precision

AUDIO OVER ETHERNET
Dante, AES67 compatible
32 channels in, 32 channels out (up to 96 kHz); 16 channels in, 16 channels out (192 kHz)
1 Gb/s Ethernet, 2 ports, transformer-balanced

INPUTS
Mic/Line inputs: 16 total, all fully featured; 6 on full-size XLR, 2 on TA3, 8 on TA5
Mic-level inputs: (XLR, TA3, TA5): Class-A, discrete differential long-tail pair, 4k ohm input impedance
Line-level inputs: (XLR, TA3, TA5): active-balanced, 4k ohm input impedance
48V phantom: full 10mA to all 16 inputs simultaneously
AES3 or AES42 available on XLR inputs 1 and 6
AES42: +10 V, 250 mA available, mode-1, auto-ASRC
Rtn A, B, C (3.5 mm/10-pin): unbalanced 2-channel, 4k ohm input impedance
Com Rtn 1,2 (TA3, 3.5mm) balanced, 1-channel, 8k ohm input impedance
External Slate Mic (TA5): balanced, 8k ohm input impedance, menu-selectable 12 V phantom

MAXIMUM INPUT LEVEL
Mic: +8 dBu (2.0 Vrms)
Line: +28 dBu (19.5 Vrms)
Rtn A, B, C: +18 dBu (6.2 Vrms)
Com Rtn 1, 2: +24 dBu (12.3 Vrms)
External Slate Mic: +12 dBu (3.2 Vrms)

HIGH-PASS FILTERS
Adjustable 40 Hz to 320 Hz, 18 dB/oct. 1st stage analog (before preamp), 2nd stage digital.

LIMITERS
Limiters available at all channels, buses, headphones, for all sample rates
Analog first stage, all subsequent stages digital
Attack time: 1 ms
Release time: adjustable, 50 ms to 1000 ms
Threshold: adjustable, -2 dBFS to -12 dBFS
Selectable knee: hard or soft
Selectable ratio: inf:1, 20:1

DELAY
Channel Adjustable 0-50 ms
Output Adjustable 0-500 ms

MAXIMUM GAIN
Trim stage (mic input): 76 dB
Trim stage (line input): 36 dB
Fader stage: 16 dB

Bus stage: 16 dB
Headphone stage: 20 dB
Mic-to-Line: 108 dB
Mic-to-Headphone: 112 dB
TA5 (along with mic input pins) for single connection to headset + mic
High output, 4 ohm output impedance, 400 mW + 400 mW at each connector, all individually driven
Compatible with headphones of any impedance

OUTPUTS
XLR (L, R) active-balanced, 250/3.2k/120 ohms (mic/-10/line)
Hirose 10-pin (L, R) active-balanced, 250/3.2k/120 ohms (mic/-10/line)
TA3 (X1-X6) active-balanced, 250/3.2k/120 ohms (mic/-10/line)
3.5mm (X7, X8): unbalanced, stereo, 1.8k ohms

HEADPHONE OUTPUTS
¼”, 3.5 mm
TA5 (along with mic input pins) for single connection to headset + mic
High output, 4 ohm output impedance, 400 mW + 400 mW at each connector, all individually driven
Compatible with headphones of any impedance

MAXIMUM OUTPUT LEVEL
(all into 10k load)
Line: +20 dBu (7.8 Vrms)
“-10”: +6 dBu (1.5 Vrms)
Mic: -20 dBu (0.078 Vrms)
X7/X8 Out: +6 dBu (1.5 Vrms)
Headphone outputs (¼”, TA-5, X9/X10): +14 dBu (4.0 Vrms)

A/D CONVERTERS
32-bit, 120 dB, A-weighted dynamic range typical
Sampling rates 44.1 kHz, 47.952 kHz, 48 kHz, 48.048 kHz, 88.2 kHz, 96 kHz, 192 kHz

DIGITAL OUTPUTS
AES3 transformer-balanced, in pairs; 1-2 (XLR-L), 3-4 (XLR-R), 5-8 (Hirose 10-pin A)
110 ohm, 2 V p-p, AES and S/PDIF compatible

RECORDING
Internal 256 GB SSD; two removable SD Cards. 10% over-provisioned for optimum performance
Simultaneous recording to internal SSD and the two SD cards
exFAT formatting
36 tracks (32 iso channels, 4 buses)
Broadcast WAV monophonic and polyphonic file format
64-bit WAV (RF64) monophonic and polyphonic; support for files > 4 G

USB
USB-C (USB 3.1 type 1) for file transfer of internal SSD, both SD Cards.
USB-A host for keyboard, external controller, external USB hubs supported for connecting multiple devices.

TIMECODE AND SYNC
Modes Supported: Off, Rec Run, Free Run, 24h Run, External, including External Auto-Record and Continuous modes.
Frame Rates: 23.98, 24, 25, 29.97 DF, 29.97 ND, 30 DF, 30 ND
Sample/Timecode Accuracy: 0.1 ppm (0.25 frames per 24 hours)
Timecode Input: 20k ohm impedance, 0.3 V - 3.0 V p-p (–17 dBu - +3 dBu)
Timecode Output: 75 ohm impedance, 5 V p-p (+12 dBu)
Word Clock Input: 10k/75 ohm selectable impedance, 1-5 V p-p input sensitivity
Word Clock Output: 75 ohm impedance, 5 V p-p output, at SR

REMOTE CONTROL
USB MIDI MCU Control - supported 3rd party fader controllers
SD-Remote Android app
USB Keyboard
External Timecode Record Trigger

LCD
320x240, Transflective, excellent sunlight visibility
Larger touchscreen display available via USB-connected SD-Remote app

POWER
External: dual 10-18 V inputs on locking TA4 connectors, pin-4 = (+), pin-1 = (-)
Dual rear-mount Sony-style L-mount batteries with chargers
Idle Current Draw: 875 mA @ 12 V (10.5 W)
Intelligent power-down of unused mic preamps and other internal circuits

ENVIRONMENTAL
Operating: -20° C to 60° C, 0 to 90% relative humidity (non-condensing)
Storage: -40° C to 85° C

DIMENSIONS (H X W X D)
5.1 cm x 32 cm x 20.5 cm;
(2.0 in. x 12.6 in. x 8.1 in)

WEIGHT
5.8 lbs (unpackaged, without batteries)
2.63 kg (unpackaged, without batteries)

FCC & ISED Compliance Statements

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Changes or modifications not expressly approved by the manufacturer could void the user's authority to operate the equipment.

This device contains transmitter module FCC ID: XF6-M7DB6
This device contains transmitter module IC: 8407A-M7DB6

FCC Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

—Reorient or relocate the receiving antenna.
—Increase the separation between the equipment and receiver.
—Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
—Consult the dealer or an experienced radio/TV technician for help.

FCC & ISED User Statement

This device complies with FCC and ISED RF exposure limits for general population / uncontrolled environments.

Cet appareil est conforme à la norme FCC et ISED les limites d'exposition pour la population générale / l'exposition incontrôlée.

A separation distance of at least 20cm must be maintained between the antenna and all persons. This device must not be co-located with any other antenna or transmitter.

This device (containing FCC ID: XF6-M7DB6, IC: 8407A-M7DB6) has been approved to operate with the antenna type listed below:

Model: GW.71.5153	Type: 2.4/5.8GHz Dipole Antenna
Manufacturer: Taoglas	Max. Gain: 3.8dBi (2.4GHz), 5.5dBi (5.8GHz)

No change to the antenna type is permitted. Any change to the antenna could result in the device exceeding the RF exposure requirements and void the user's authority to operate the device.

This Device complies with Industry Canada License-exempt RSS standard(s). Operation is subject to the following two conditions: 1) this device may not cause interference, and 2) this device must accept any interference, including interference that may cause undesired operation of the device.

Cet appareil est conforme avec Industrie Canada, exempts de licence standard RSS (s). Son fonctionnement est soumis aux deux conditions suivantes: 1) ce dispositif ne peut pas causer d'interférences, et 2) ce dispositif doit accepter toute interférence, y compris les interférences qui peuvent causer un mauvais fonctionnement de l'appareil.

Incorrect use of batteries poses a danger of explosion. Replace only with the same or equivalent type. Properly recycle batteries. Do not crush, disassemble, incinerate, dispose in a fire or expose batteries to high temperatures.

Glossary

¼-inch jack
Common analog audio connector used as both an audio input and output. When a ¼-inch jack is described as TRS (tip-ring-sleeve) it can be wired as either a balanced connection or as a two-channel connection. ¼-inch headphone jacks are typically wired as TRS stereo jacks.

3.5 mm jack
common small-format audio connector. Often used for headphones and -10 dBV signals for portable audio devices.

10-pin Hirose
A high-density multi-pin connector commonly used to connect an audio mixer with video cameras. Sound Devices has numerous field mixers with 10-pin Hirose connectors. Dedicated fantail breakout cables from Sound Devices and other third parties offer Hirose-to-XLR and 3.5 mm connections to simplify connection.

AES3
a standard for the exchange of digital audio signals between professional audio devices. An AES3 signal can carry two channels of PCM audio over balanced, 110 ohm interconnections. AES3 is most commonly interconnected with XLR-3 cables.

AES42
a digital interface protocol for microphones and microphone inputs. Microphones conforming to this standard directly output digital audio through an XLR or XLD male connector, rather than producing an analog output. AES42 microphones require powering.

attenuation
A reduction in the level of an audio signal. Attenuation can be applied to both analog and digital signals. A fader is used primarily to attenuate signals, though a small amount of positive gain is often available on a fader.

bext chunk
Broadcast WAV extension data added to the audio data in a WAV file. The bext chunk includes timecode and user bit data. For systems that do not recognize the bext chunk this additional information is ignored.

bias voltage
voltage typically applied to a lavalier microphone from a wireless transmitter or XLR adapter to power condenser capsules and/or impedance converters. Different from phantom power, bias voltage is most commonly single-ended, being sent only on one connection.

bit depth
When converting between analog and PCM digital audio the amplitude of an analog signal is measured in finite steps, measured in bits. Higher bit rates result in greater resolution of amplitudes, resulting in higher dynamic range. 24-bit audio, with a theoretical maximum dynamic range of 144 dB, is the standard bit depth used throughout the audio chain for production.

broadcast WAV, BWAV
Broadcast WAV files are WAV files with additional, non-audio data, such as bEXT chunk data. Broadcast WAV files offer timecode support.

Bus
An audio path that is the destination of one or multiple (mixed) channels. A bus is typically routed to an output, a record track, or both.
camera return
An audio input on a mixer designed to receive the output, typically the headphone output, of a camera. Camera return inputs allow the user to monitor the level and quality of the signal received at the camera. In Scorpio the camera returns can be used as a source for any channel.

channel
A “slot” of a mixer that is controllable and routable. A given input feeds the channel and the channel’s settings process and route the audio as required. It can also be thought of as the path its selected input signal takes on its way to its record track, a bus, or an output.

channel grouping
With Scorpio any of the first 16 channels can be grouped together so that their faders, record arming state mute states can be controlled together. Channel grouping can be used as an alternative to sending channels to a bus.

circled take
An identifying character, the @ symbol, which is placed in the file name to highlight a take. Circled takes can either be used to identify good takes or to identify tracks or takes that will be ignored.

com return
A dedicated audio input designed to receive signals from a PL, or private line communications circuit. The com return on Scorpio can routed to an output or a bus.

com send
A dedicated output designed to send signal to a PL (private line, talkback) communications circuit. The com send is toggled by a front panel switch.

Dante
a combination of software, hardware, and network protocols that deliver uncompressed, multi-channel, low-latency digital audio over a standard Ethernet network using Layer 3 IP packets.

dBFS
A measurement of the signal level of a digital signal in dB increments, dB relative to full scale signal. The maximum signal in dBFS is 0 dBFS, with signals expressed with a negative sign. dBFS signal strength is an internal measurement and does not correspond to analog signals unless the relationship between analog signal and digital signal is known.

delay (channel)
Time delay that can be applied to an individual channel. Channel delay, typically set in milliseconds, is often used to compensate for different acoustical or electrical arrival times of signals between channels.

Ethernet
A family of computer networking technologies. Ethernet commonly refers to the physical interconnection of the network typically using twisted-pair copper connections on CAT cable with RJ-45 connectors. Common Ethernet data speeds include 10 Mbs, 100 Mbs, and 1000 Mbs.

exFAT
A storage volume format that can be read and written from current versions of MacOS and Windows. exFat supports volume sizes up to 128 PB (gigantic), and individual files can have a maximum size of 16 EB (even more gigantic, bigger than the maximum volume size).

fader
A physical control on a mixing console, either a rotary or sliding potentiometer, that controls the level of a channel to a bus. Most faders have more attenuation than gain available and a unity gain position where the input trim level established the level to the bus.

false take
A recorded take that was either erroneously recorded, or a take that needs to be repeated. It can be labeled after recording. An identified false take is moved to the trash bin and the auto-incrementing take number is reset to the value prior to the false take.

file list
Every file recorded by a recorder is visible in the file list. It can be viewed either on a recorder or from a computer when the recording volume is mounted. The file list shows all the individual files recorded by a recorder.

frame rate
The rate at which video or motion picture images are recorded or played back, measured in frames-per second (FPS). All audio and video devices must be running at the same frame rate to keep audio and video synchronized. Timecode frame rates are either an integer or non-integer value. Integer values include 24, 25, and 30 FPS. Non-integer frame rates include 23.976 and 29.97, and 29.97 drop FPS.

frequency
The period at which a wave oscillates, measured in hertz (Hz). Frequencies audible to humans range from 20 Hz for very low frequency signals to 20 kHz for very high frequency signals.

gain
An increase (or decrease with negative gain) in the level of an audio signal. Gain can be applied in several locations, to both analog and digital signals. In a field mixer the microphone preamplifier provides a substantial amount of gain at the trim to raise the low level microphone signal to a usable signal in the mixer. Gain is also available at the fader. Gain of digital signals or line level analog signals is often limited. Unity gain is gain stage that neither adds or subtracts level from a signal.

headphone monitor
Often a separate bus with a dedicated headphone volume control, the headphone monitor typically is normalled to the main left/right output bus of a mixer. Headphone sources can often be selected among soloed tracks or buses. In some products complex headphone monitoring of MS Stereo, LR stereo, and ambisonic sources is available.

high pass filter (audio)
Also referred to as a low-cut filter, this circuit reduces the amount of low frequency content in an audio signal. A HPF is particularly useful when recording speech since the human voice does not generate appreciable energy at low frequencies. The HPF reduces non-speech signals such as environmental noise, wind noise, and microphone handling noise, improving the intelligibility of speech and reducing low frequencies from overloading the input. The high pass filter is placed in the circuit close to the microphone preamplifier.
High pass filters are often frequency selectable, ranging from 20 Hz to 200 Hz. HPF also have a slope, generally from 3 dB/octave to 18 dB/octave. Greater/steeper slopes offer more attenuation of frequencies just below the set filter frequency.

input
The physical connection and associated signal type from external sources connected to a device. Inputs can include microphone inputs on XLR connectors, Dante inputs on audio-over-Ethernet, and USB audio inputs from a computer. Depending on the architecture of the mixing console its inputs may be hardwired to channels or channels can be selected from different inputs.

input limiter
A limiter circuit reduces the peak signal levels of audio, generally to prevent signal overload. Analog inputs have a maximum input signal level that can be reached before overload/distortion is introduced. Setting the input gain correctly so that input signals do not reach this maximum level prevents most overload conditions. In the presence of very high, unexpected signals an input limiter changes the gain of the incoming signal and prevents it from overloading. Input limiters are sometimes compressor-type circuits with a ratio of infinity:1, meaning that any increase to the input signal into the limiter at the limiter threshold does not increase the output signal of the limiter.
Several parameters may be available in a limiter, including knee, ratio, release, and threshold.

isolated track
A recorded track of an individual microphone or sound source. “Iso” recordings allow for post-record mixing of individual sound elements.

iXML
An extensible data schema for audio and related metadata stored in broadcast WAV files. Manufacturer-specific data generated during recording is stored in iXML.

line level
an analog audio signal used to interconnect audio equipment. Line level may be balanced or unbalanced, referenced to +4 dBu or -10 dBV, professional or consumer respectively.

low cut filter
See high pass filter.

microphone level
the audio audio signal generated by a microphone. Mic level signals are very low level, requiring a microphone preamplifier to bring them to usable, line levels. Interconnects with microphone level signals can be subject to noise and interference.

mid-side linking (inputs)
When mid-side (MS) stereo inputs are used and the inputs are set to MS linking and MS decoder is activated for those inputs. This yields a stereo signal with one fader controlling overall input level and the other fader controlling the "width", or amount of the side signal added. With an MS matrix at the input, the signal is sent to an output bus as left/right stereo. Mixers with MS matrices often allow for discrete mid and side signal recording. In that case the MS decoder can be activated at the headphone selection to monitor left/right stereo.

mix track
A recorded track that is a sum of multiple tracks. In production sound the mix track is often a single summed track of all production dialog elements. Mix tracks can also be sub-mixes of like microphones, such as a sub mix of just lavalier microphones or just boom microphones.

monophonic WAV
A WAV file that is comprised of a single track of audio. When recording multi-track audio with monophonic WAV files each track is recorded to its own WAV file, with a file name indicating the track number. All associated monophonic files that are part of a multi-track recording will be identical lengths.

mute
A mute control is a convenient on/off switch for a channel and an easy way to remove a channel from appearing in downstream buses. Mute an input or channel does not change levels or settings; when channels are muted and unmuted, their settings remain.

notes (metadata)
A metadata field that is saved along with audio data in a recorded sound file, useful for sound report generation. Some workstation software recognizes the notes field and presents it when viewing the sound file.

output
The physical connection and associated signal type sent from a device. Outputs can be source from inputs, buses, record tracks, and other auxiliary signals.

output auto-mute
When set, an output signal is muted when recording is stopped, restricting program audio from being sent to listeners "between takes".

output delay
A digital delay applied at the output. Signal delay is often set at an output to compensate for the delay introduced by digital imaging systems so that picture and sound remain in correct "lip sync". Output delay is set in either frames or milliseconds.

pan
When a channel is routed to a stereo-linked bus the level it appears at each bus is adjusted by a pan control. A channel with its pan control "straight up the center", or "centered" sends signal at the same level to each bus. A channel that is panned left or right sends the signal to the left or right bus, respectively.

PFL, pre-fade listen
When an input or channel is selected for monitoring/solo with a PFL, the channel is routed to the headphone output before the channel fader so that the fader position has no effect on the headphone level. Trim/gain changes to the input will change the headphone output.

phantom power
Condenser (capacitor) microphones require power for operation. They use power to charge the diaphragm backplate (for true condensers) and power the impedance convert located adjacent to the microphone capsule. Phantom power is the method for microphone inputs to supply DC power to the microphone through the same connection used for the audio signals from the microphone.

Phantom power provides a positive voltage, typically between 11-52 VDC, with 48 V being the most common, on both pin-2 and pin-3 with pin-1 used as ground. The DC voltage appears as a common-mode signal on the balanced connection and is rejected by the connection's differential amplifier. Phantom power has no effect on dynamic microphones.

phase
The relationship one audio signal has in time with respect to another audio signal, defined in degrees of phase. When audio signals are generated at identical times, they are "in phase" with each other. When one audio signal is time-delayed with respect to another the signals are "out of phase". Differing phase relationships can be introduced several ways, including when microphones are placed at varying distances from a sound source, or electrical/digital delay is introduced to one signal with respect to another.

phrase
A pre-set text string which can be used to quickly fill out the notes field.

pre-fader routing
The signal from a channel is routed to a bus before the fader in the signal path. The input trim, if available, controls the channel level sent to the bus. Isolated tracks are typically recorded pre-fader so that any level changes made to the faders don't affect the recorded signal.

pre-roll
A continuous buffer that is always writing to memory offering a recording that begins prior to when the record button is activated. Pre-roll is set in seconds, and the recording begins the set number of seconds prior to the button being pressed. This is helpful in applications where an operator missed a cue to begin recording.

polarity (audio)
The direction of the current flow of an audio signal is defined as polarity. The polarity of a signal can reversed when a balanced audio signal connection has its pin-2 and pin-3 connections reversed. Single-ended signals can have their polarity reversed when going through an "inverting" gain stage. It is best practice to have all incoming and outgoing signals with the same polarity relationship.

polyphonic WAV
An individual WAV file that contains multiple audio tracks. When recording multi-track audio with polyphonic WAV files all recorded tracks are contained within a single WAV file.

post-fader routing
The signal from a channel is routed to a bus after the fader in the signal path. The fader controls the level of the channel at the bus. Channels sent to a master bus, such as the left/right bus, are typically sent post-fader.

post-roll
An extra period of time that is appended to the end of a recording when stop is pressed. If record is pressed during this period of time, recording will resume within the same file with no audio lost. This is particularly useful should a Director call 'cut' prematurely or accidentally.

project
An option available for file organization on Sound Devices recorders. Projects are the highest level of file folder organization. The project folder can contain sub-folders of scene files or recorded files directly.

record bell
A tone generated in headphones to alert the listener that recording has started. The bell is also produced when recording has ended with the stop button, when the recording volume is full, or when power is in a critical state.

sampling rate
When converting between analog and PCM digital audio the analog signal is measured (sampled) in unique steps at a data rate specified in kHz. Higher sampling rates allow for representing higher frequency analog audio. 48 kHz is the standard sampling rate for production, worldwide. Higher sampling rates including 96 kHz and 192 kHz are used for high-precision applications where the representation of audio above 20 kHz is required. A general rule is that the maximum analog audio frequency is ½ the sampling rate.

scene
On Sound Devices recorders the scene becomes part of the file name for a take. Scene names can be pre-loaded to quickly change between scenes.

slate microphone
A microphone, built-in or external microphone, on an audio mixer used to notate takes or communicate with sound team members by the mixer's user speaking into the microphone. Slate microphones are often routable to buses or tracks.

stereo linking (inputs)
When active for stereo sources such as stereo microphones, linked inputs are hard panned to the left and right bus. Controls including gain (trim), fader, high pass filter, delay, limiter, mute, and routing are controlled together.

solo
A control on a mixer to route a channel to headphones while muting all others. Solo and PFL are related controls and in many consoles are the same. Solo circuits can be exclusive—only one channel is sent to headphones at a time—or non-exclusive—any number of channels can be sent to the solo circuit and appear in headphones.

TA-type connector (TA3, TA4, TA5, TA6)
Miniature XLR-type, locking connectors. TA3 connectors are used by Sound Devices for various inputs, outputs, and as balanced and unbalanced connections. TA4 connectors are used by Sound Devices for DC power connections to the Scorpio mixer-recorder. TA4 is also used for audio connections from lavalier microphones to some wireless transmitters. TA5 and TA6 connectors are presently not used by Sound Devices though they are used for audio connections by other manufacturers.

take
A recorded take is an individual recorded file (or files when recording monophonic WAV files) generated by a recorder. Take numbers are auto-incrementing. Take numbers are added to the end of the file name.

take list
Separate from a file list, a take list consolidates related files such as a group of monophonic WAV files generated by a single take and presents them as a single take.

test tone
See tone oscillator.

timecode
A numerical clock value expressed in hours:minutes:seconds:frames, i.e. 04:59:39:05, used to synchronize cameras, video decks, and audio recorders. Timecode requires clocks on devices to be synchronized, either through a wired or wireless connection between devices, or through a process called "jam synch" where each device, which requires a high-precision clock, runs independently after their clocks are synchronized.

timecode mode
Sound Devices recorders offer multiple timecode modes. Different modes correspond to different timecode workflows. Common modes available in Sound Devices recorders include:
record run - timecode advances only when recording is engaged
free run - timecode run continuously, typically with the start of production being at 0 hour
24 hour - similar to free run except the start time corresponds to time-of-day
ext TC - the recorder applies the value of an external timecode source.

tone oscillator
A sound generator producing a sine wave tone at a given frequency at a given output level. With its known output level tone oscillators are helpful to set gain structure between audio equipment.

track
A single recorded audio signal. Common recorded tracks are the main left/right master audio bus and isolated (iso) channel recordings. Iso tracks are typically identified by the channel of the same number, e.g. channel 1 is sent to track 1, channel 2 is sent to track 2, etc.

track arm
Tracks that are active and ready for recording are said to be "armed". When recording begins all armed tracks begin recording. Depending on the production is may be advantageous to arm and disarm tracks, especially to disarm unused tracks.

track name
Individual tracks of a multi-track recording can be named to indicate microphone type or character name.

trim
Also defined in mixers as "gain", the trim adjustment is the first stage of gain of a microphone or line level input. Typical microphone trim values range from 10 dB to 50 dB, depending on microphone sensitivity and volume of the sound source.

user bits
Static, numeric data that is available as part of a timecode signal. User bits are often used to indicate the date of a file. User bits are four sets of two-digit hexadecimal numbers from 00 to ff.

WAV File
A universal, well-supported file type for sound file recordings. WAV files can contain one or more (up to 65,535) tracks of PCM audio data at any sampling rate and bit depth. A standard WAV file is limited to a maximum file size of 4 GB. Sound Devices uses the .WAV extension for recorded files, including for files with Broadcast WAV metadata and WAV RF64 files.

WAV RF64
An extension of the WAV file type that supports file sizes larger than 4 GB. When recording high track count, high sampling rate polyphonic WAV files, the 4 GB size limitation of WAV can be reached quickly. RF64 files larger than 4 GB require recording to a volume type than can support file sizes larger than 4 GB.

word clock
A reference signal used to synchronize the sampling rate of multiple digital devices.

XLR female
Industry-standard 3-pin locking audio connector for microphone and line-level sources. Predominantly used as an input. Also shown as XLR-F

XLR male
Industry-standard 3-pin locking audio connector for microphone and line-level sources. Predominantly used as an output. Also shown as XLR-M

Post Office Box 576
E7556 State Rd. 23 and 33
Reedsburg, Wisconsin 53959 USA

support@sounddevices.com

+1 608.524.0625 main
+1 608.524.0655 fax
800.505.0625 toll free

www.sounddevices.com